E1010

OPERATING MANUAL

YAMAHA

Introduction

The E1010 Analog Delay Line delivers a full 300 milliseconds of time delay, plus special effects. It also delivers consistent audio performance, not only at the shortest delay time, but even at the maximum of 300 milliseconds — whether the input level is low or high. This is because the E1010 combines quiet, low-distortion amplifiers with a carefully engineered compandor to reduce noise. Thus the output is always clear and natural sounding.

What is an analog delay line? It is a device that provides echo and related effects by slowing down audio signals — a more capable space-age replacement for older tape-loop echo machines. The term "analog" means that the audio signal retains its original voltage levels throughout the electronics. It is not converted to binary numbers, "digitized," as is done with "digital" delay lines. Hence there is no "quantizing" or "digitizing" noise.

All controls are clearly labeled and are recessed to avoid inadvertent changes in settings (recessed controls are also safer in shipping and cartage). There are rear-panel input and output jacks for rack-mount installations, plus front panel jacks for convenience in portable applications. There is even a foot switch jack so the performer can switch effects IN and OUT while playing. Guitar players will especially appreciate the E1010's very high input impedance and its buffered input level control which together prevent pickup loading and thereby preserve the instrument's frequency response. Overall, the E1010 Yamaha offers tremendous flexibility, but not at the expense of sound quality. The E1010 is an economical and sensible choice for creative soundmen and vocal or instrumental performers. Because the E1010 is completely solid state, there are no tape loops to replace, no heads to adjust and clean, and no mechanical transports to service. In fact, no routine maintenance is required; in the rare event of a problem, the E1010 is backed by Yamaha's extensive service facilities. The unit measures just 3-3/4" (96mm) high x 19" (480mm) wide x 9-1/2" (243mm) deep (rack mountable) and weighs only about 10 pounds (4.7 kg), and is well suited to portable or fixed applications.

This instruction manual was prepared to assist you in getting the most out of your E1010. While you may already have begun using the unit, we urge you to read this manual thoroughly, and to re-read it as you become more familiar with the E1010's features and functions.

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NOTE: There are no user serviceable parts inside the E1010. See caution on inside back cover of this manual.

Brief operating instructions

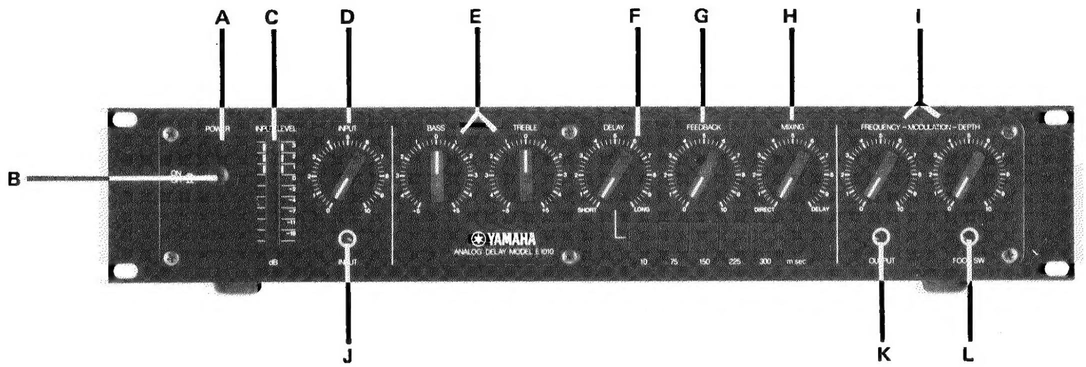


Fig. 1 — Front Panel

A. POWER INDICATOR
This LED (Light Emitting Diode) is illuminated whenever the AC power is switched ON.

B. POWER SWITCH
This recessed latching pushbutton turns the E1010 AC Power ON and OFF. Transient supression circuitry avoids loud turn-on or turn-off "thumps."

C. INPUT LEVEL DISPLAY
The display monitors the signal level after the E1010 Input level control. The display has 9 LED's which indicate peak values from −18dB to +3dB (0dB = −30dBm).

D. INPUT LEVEL CONTROL
This control adjusts the input sensitivity for optimum interface (i.e., for low noise and low distortion). Set the control so the display indicates "0dB" on program peaks, with average levels of about "−11dB."

E. BASS & TREBLE TONE CONTROLS
These controls are provided to alter the characteristics of the delay line—not just the treble/bass content, but the overall effect. The Bass and Treble controls modify only the delayed signal, not the direct sound. Each control provides a maximum 12dB of boost or cut, with turn over points at 70Hz (Bass) and 7kHz (Treble).

F. DELAY SWITCHES & CONTROL
Five pushbuttons select different ranges of delay time, in milliseconds: 3-10ms, 25-75ms, 50-150ms, 75-225ms, and 100-300ms. An adjacent Delay control enables the actual delay time to be continuously varied from the minimum to the maximum within each range.

G. FEEDBACK
This control adjusts the proportion of delayed sound from the E1010 output that is mixed back into the E1010 input. Increasing the amount of feedback lengthens the duration of an echo, and adds sustain to other effects as well. If Feedback is set high enough, regeneration can occur (oscillation or run-away echo). This can be cured by lowering the Feedback setting or by varying the Delay time up and down.

H. MIXING
The E1010's front and rear panel Output jacks can be varied from all delayed to all direct sound. Full counterclockwise rotation of the Mixing control yields all direct sound; as the control is rotated clockwise, more delayed sound is mixed in until the Output jacks carry all delayed sound. The rear-panel Direct Only output jack always has 100% direct sound and is not affected by the Mixing control.

I. MODULATION FREQUENCY & DEPTH CONTROLS
Within a given delay range, the actual delay time can be varied manually with the Delay control, or it can be varied automatically with these Modulation controls. The Depth knob sets the amount of deviation in delay time. The maximum deviation increases in longer delay ranges. The rapidity of the deviation in delay time (the speed of the effect) is set with the Frequency knob. Modulation is useful for vibrato, flanging, phasing, and similar effects.

J. INPUT JACK
This standard phone jack is unbalanced and has a nominal −30dB (24.6mV) sensitivity. (The impedance is very high, 800kohms.) Another Input jack is provided on the rear panel, but this front-panel Input jack takes precedence; if sources are connected to both inputs, only the front input will be fed to the unit.

K. OUTPUT JACK
This standard phone jack is unbalanced and has a nominal −20dB (78mV) output level. It is intended for driving high impedance inputs. This jack is parallel wired with the rear-panel Output jack, so both may be used at the same time. This output will provide delayed and/or direct sound depending on the position of the E1010's Mixing control.

L. FOOT SWITCH JACK
This standard phone jack is provided for use with a guitar-type foot switch. The switch turns all delay effects ON and OFF (tone, delay time, feedback and modulation), but it does nothing to the direct sound. A foot switch is handy for initiating an echo effect at the end of a vocal or instrumental phrase, for abruptly ending an effect on a precise cue, etc.

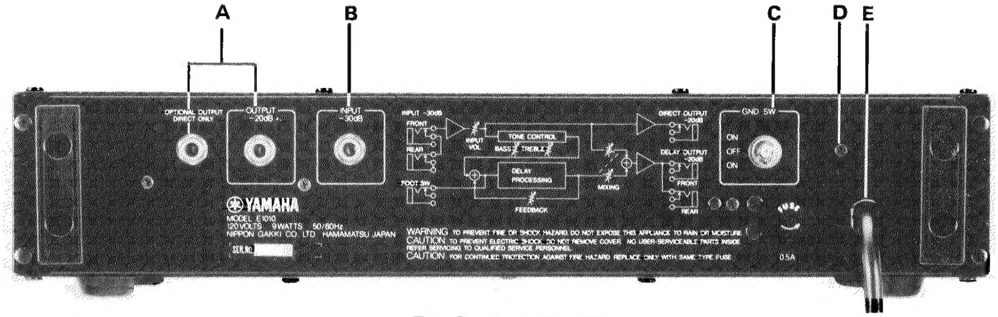


Fig. 2 — Rear Panel*

A. OUTPUT JACKS

Separate Output and Direct Only Output phone jacks are provided, both having nominal output levels of −20dB (78mV). The jacks may be used at the same time as the front-panel Output jack. Depending on the setting of the E1010's Mixing control, the rear panel Output jack will provide all delayed, all direct, or any blend of sound. The Direct Only output carries the same signal as applied to the E1010 Input, subject only to the Input Level control.

NOTE: Some effects can be achieved only when the direct and delayed sounds are electrically mixed, either within the E1010 via the Mixing control, or externally if the E1010's Mixing control is set fully clockwise (to all "Delay" sound). Other effects can be achieved only when the direct and delayed sounds are kept separate electrically. If an expected effect is not achieved, be sure the appropriate output is used with a suitable Mixing control setting.

B. INPUT JACK

This jack is identical to the E1010's front panel Input jack. The front panel Input takes priority, however, so the rear panel Input will be disregarded as soon as anything is plugged into the front panel Input jack.

C. GROUND SWITCH (For U.S. and Canadian models only)

This three position switch may reduce noise or hum in some installations. Use the switch position which sounds best. OFF does not interrupt the ground path between the AC cord and the chassis (it merely removes a filter capacitor from the AC line).

D. FUSE

This 0.5 amp fuse for nominal 120V AC mains on U.S. and Canadian models or 250mA fuse for nominal 220V AC mains on other models protects the delay line by preventing excess current flow in the AC side of the power supply.

E. AC POWER CORD

This grounded (3-wire)AC power cord may be wound around holders on the E1010 rear panel when not in use.

HINTS FOR QUICK SETUP

Plug the cable from your microphone, instrument, or mixer effects output into either of the E1010 "Input" jacks. Connect a cable from either of the E1010's "Output" jacks (not the "Direct Only Output") into the input jack of your amplifier, mixer, tape machine, etc. Then set up the E1010 for the desired effect, referring to the diagrams on Pages 8 and 9 of this manual.

If you don't hear any sound right away, check the E1010 Power switch and Input level control, and make sure that the display is active during louder passages. Be sure the input and output cables are connected properly.

Short delays are useful for flanging, double-tracking, tunnel sounds, and similar effects, Medium and long delays are useful for vibrato, echo, pitch bending, and rhythmic effects. The delay ranges have considerable overlap to ensure that modulation effects can be obtained without running into restrictions at the end of a given delay range.

The tone controls themselves operate the same as most tone controls, but they can modify the actual effect as well as the frequency balance. For example, in short delays the Bass and Treble tone controls can change the nature of the comb filter or flanging effect. In medium or long delays with feedback, the tone controls can change the effective delay time, depending on the frequency content of the input signal.

While you can always refer to diagrams, you will probably want to learn to figure setups and effects "from scratch." This is easier when you understand what the E1010 is doing to the sound. After familiarizing yourself with the unit, we recommend that you study the section of this manual titled "How the E1010 Works."

** The rear panel show here is subject to U.S. specifications.*

PRECAUTIONS

1. Before turning ON the E1010, set its FEEDBACK control to minimum (full counterclockwise). This avoids any potential howling which could damage speakers.

2. While the E1010 is protected against turn-ON and turn-OFF "thumps" or transients, it is still a safe practice to Lower the volume of your power amplifier when first turning ON the E1010. Also, begin with the E1010 INPUT control at a low setting and gradually turn it up until program peaks illuminate most of the LED's in the unit's INPUT LEVEL display. Then adjust the power amplifier volume as desired.

3. Use caution whenever connecting the E1010 output to its input; mixing and/or feedback can be achieved without special connections by using the controls provided.

4. Never connect the output of a power amplifier to the E1010 inputs; they are designed for mic, instrument or line level signals only (Exception; a guitar amp output may be used if it is connected via a suitable direct box, one including an attenuation pad to lower the signal to mic or line levels.)

5. Do not connect power supplies or apply DC voltages to any E1010 inputs or outputs.

6. Never use abrasives or spray-on chemical solvents to clean the E1010. Mild detergents applied to soft cloths are preferred.

General specifications

Delay Time (Continuously variable within each range)
10ms range, 3 to 10ms;
75ms range, 25 to 75ms;
150ms range, 50 to 150ms;
225ms range, 75 to 225ms;
300ms range, 100 to 300ms.

Feedback
Variable, from none to 100%.

Modulation
Sine-wave variation of delay time relative to set delay.
FREQUENCY (speed) from 0.5 to 10Hz.
DEPTH from 0 to ±10% @ 10ms: max. depth increases with delay to ±30% @ 300ms.

Tone Controls (shelving curve)
BASS: ±12dB @ 70Hz; TREBLE: ±12dB @ 7kHz.

Frequency Response
DIRECT ONLY: +1, –3dB 20Hz – 20kHz.
OUTPUT DELAY: ±3dB, 30Hz – 8kHz @ 10ms;
±3dB, 30Hz – 2kHz @ 300ms.

Total Harmonic Distortion
DIRECT ONLY: Less than 0.1%, 20Hz – 20kHz.
OUTPUT DELAY: Less than 2.0%, @ 1kHz;
@ –10dB* output into 10kohms.

Intermodulation Distortion
DIRECT ONLY: Less than 0.5%.
OUTPUT DELAY: Less than 3.0%, @ 10ms delay; using frequencies of 70Hz and 7kHz, mixed 4:1, @ –10dB* output into 10kohms.

Hum and Noise**
DIRECT ONLY: –100dBm equivalent input noise, –90dB* output noise with input volume at maximum (70dB S/N).
OUTPUT DELAY: –87dB* output noise with input volume at maximum, 10ms delay; –87dB* output noise with input volume at maximum, 300ms delay.

Indicators
POWER ON: LED turns ON when Power is ON.
INPUT LEVEL: 9-segment LED display shows peak input level in dB (relative). Calibrated in steps of +3, +2, +1, 0, –2, –4, –7, –11 and –18dB.

Power Requirements
120V AC (for U.S. and Canadian models) or 220 or 240V AC (for other models) nominal, 50/60Hz, 12W.

Dimensions
3-3/4" high x 19" wide x 9-1/2" deep (96 x 480 x 243 mm); rack mountable.

Net Weight
10 pounds, 4 ounces (4.7 kg)

INPUT CHARACTERISTICS

Impedance
Buffered input, 800kohms actual load; for use with nominal 600 ohm lines, 5kohm lines, or high impedance instrument pickups.

Input Level
–30dB* (24.6mV) nominal sensitivity; +18dB* (6.2V) maximum before clipping.

Connectors
Unbalanced standard phone jacks on front and rear panels; wired so that phone plug inserted in front panel input jack internally disconnects the rear panel input source.

OUTPUT CHARACTERISTICS
(Same for Output and Direct Only Output)

Impedance
250 ohms actual source impedance; for use with nominal 10kohm or higher impedance loads.

Output Level
–20dB* (78mV) nominal; –3dB* (0.5V) maximum before clipping.

Connectors
Unbalanced standard phone jacks. Rear panel Direct output jack. Parallel-wired front and rear panel Delay output jacks.

**0dB is referenced to 0.775 volts RMS.*
***Measured with –6dB/octave filter @ 12.47kHz; equivalent to a 20kHz filter with infinite dB/octave attenuation.*

Specifications subject to change without notice.

Installation

MOUNTING

The E1010 is designed so it can be mounted in a standard 19" wide equipment rack, where it occupies a 3-1/2" panel space. Alternately, the unit may be placed on a shelf, table, atop other equipment, or in a small custom cabinet. As with all low-level signal processing equipment, it is advisable to locate the E1010 away from sources of heat or strong electro-magnetic fields (i.e., away from power transformers).

CONNECTIONS

In rack mounted installations, the rear panel input and output jacks may be utilized for "permanent" cable connections. In any case, the front panel jacks remain accessible for temporary connections to and from the E1010. All connections are unbalanced, utilizing standard 1/4-inch tip/sleeve phone jacks; high quality guitar-type cables with rubber insulation, high shield density and low capacitance are recommended.

The E1010's high input impedance permits the unit to be fed by low or high impedance outputs. Should the unit feeding the E1010 require a low impedance termination (i.e., some passive equalizers rated for 600-ohm actual loads), a suitable resistor may be installed across the signal and shield conductors of the input cable. The INPUT level control permits the E1010's sensitivity to be reduced so it will accommodate +4dB (0.775V) nominal line levels.

The E1010's outputs are designed to drive high impedance loads, 10kohms or more at nominal –20dB levels (low line level); this level and impedance is typical of a great deal of musical sound equipment. If the E1010 output is connected to a mic-level input, it is sometimes helpful to use an external 20dB or 30dB attenuation pad to avoid overdriving that input.

TYPICAL HOOKUPS

The following illustrations are typical applications of the E1010. Suggested control settings for obtaining various effects are illustrated in Figure 7 on Page 8.

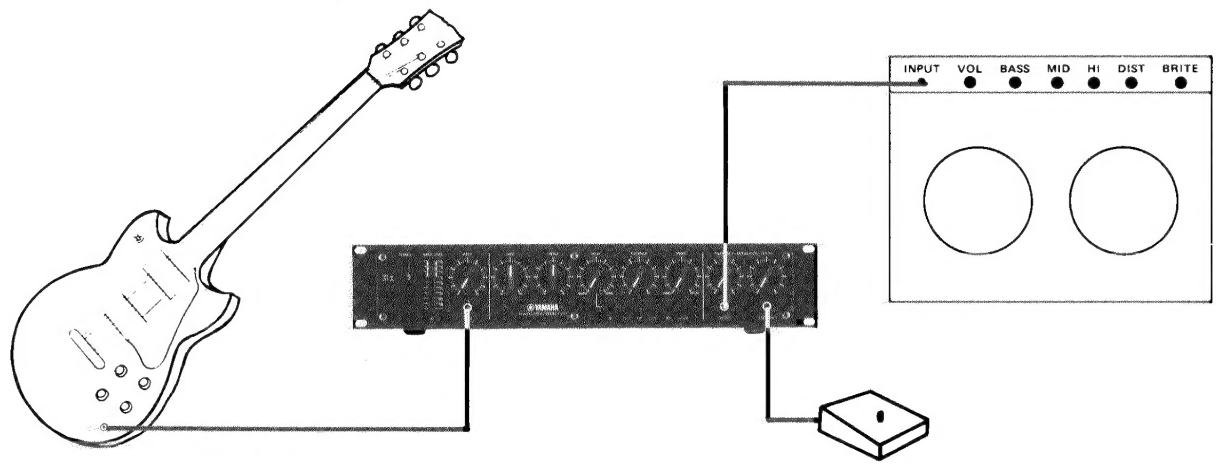


Fig. 3 — Hookup for special effects with a guitar amp that does not have PREAMP OUT & POWER AMP IN Jacks or an effects patch point.

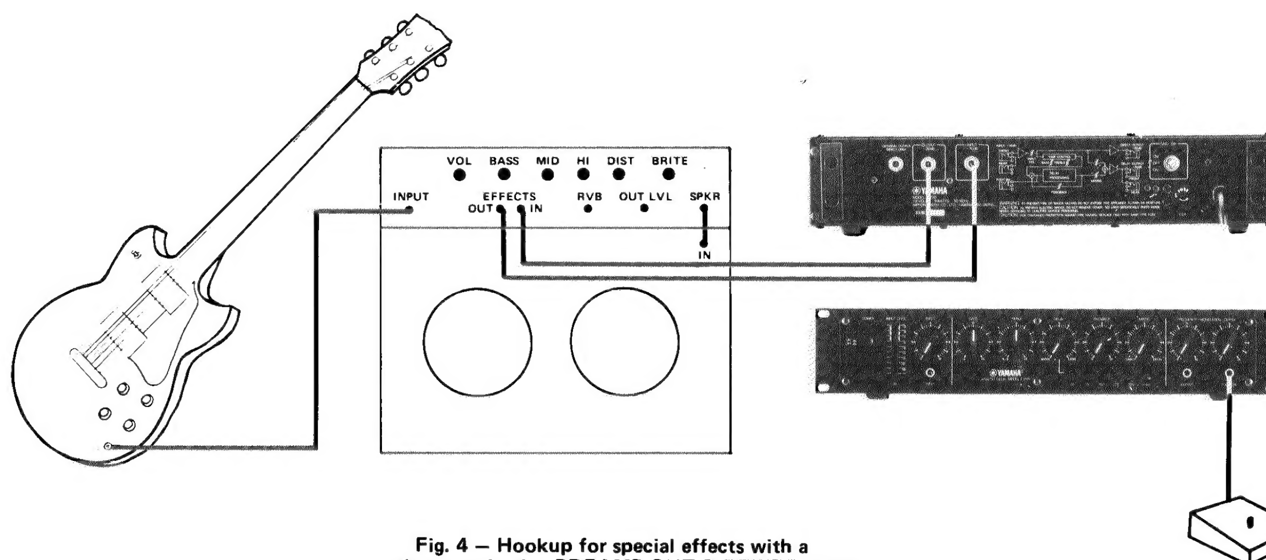


Fig. 4 — Hookup for special effects with a guitar amp having PREAMP OUT & POWER AMP IN Jacks or an effects patch point.

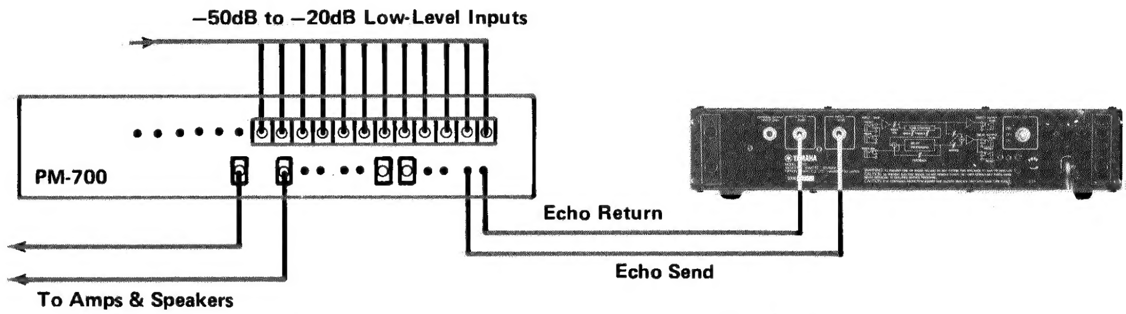


Fig. 5 — Hookup for delay or special effects with a mixer or console:
A. In An Echo Send/Return Loop

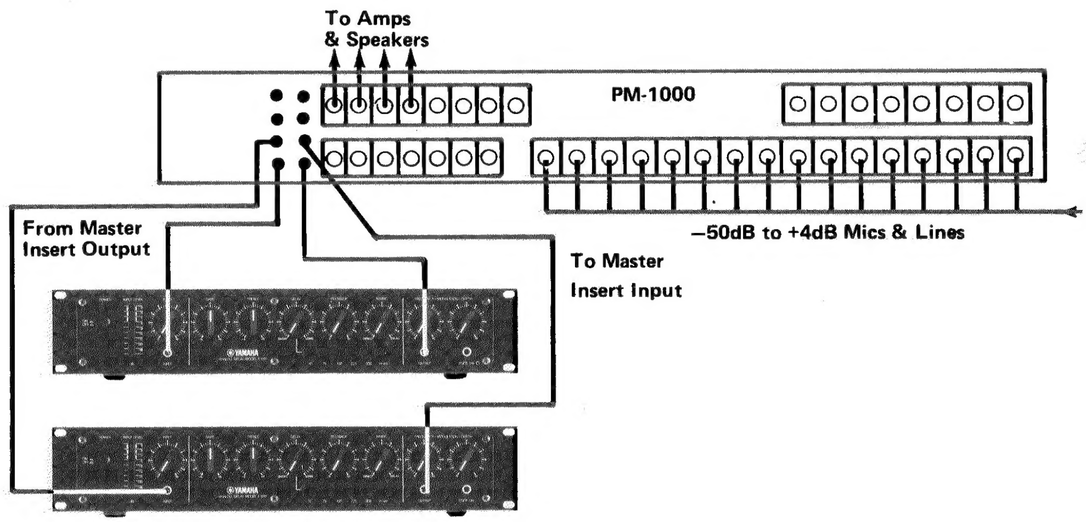


Fig. 5B — In An Insert Loop (Patch Point)

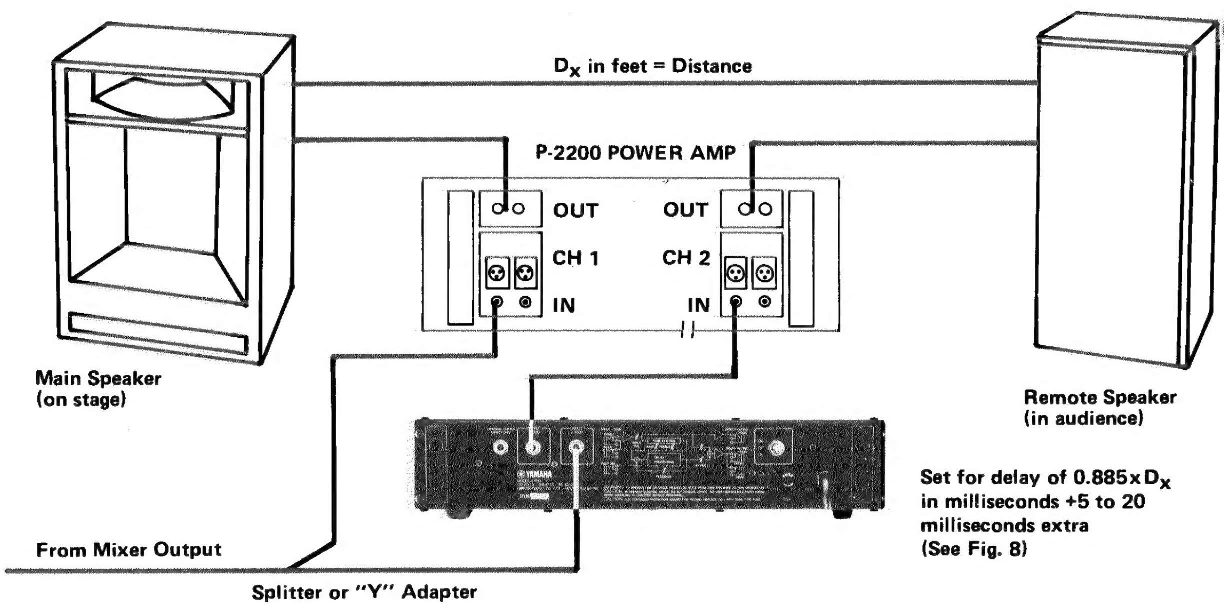


Fig. 6 — Hookup for delay of signal to the power amp of a remote speaker in a distributed sound reinforcement system.

Applications

See next page for E1010 Front Panel Settings and Musical Representations of Various Effects.

Stereo From A Mono Source

Use the E1010 to delay the sound to one of two speakers by approximately 30 milliseconds. This delay is too short for the listener to perceive an echo, but is long enough to spread the image, simulating natural ambience. This type of stereo spread is not recommended when the program is also being combined to a mono signal.

Double Tracking

Double tracking can be obtained by overdubbing the exact same instrumental or vocal part on a second track of the tape. The sound achieved is never in perfect unison due to minor variations in pitch and time. Double tracking is a very desirable and useful effect (also known as chorusing), especially for richening the sound of vocals, strings, and other instruments.

The E1010 creates this effect in real time, allowing you to precisely control the variables (pitch and time deviations). The delay time is set for about 40 milliseconds and is automatically increased and decreased by the E1010's built-in Modulation controls (Frequency and Depth).

Echo & Reverb Effects

"Echo" and "reverb" are often confused with one another, and there is some overlap in their definitions. Generally speaking, "echo" consists of one or more distinct, delayed sound images with recognizable attacks. "Reverb" also consists of multiple delayed sound images, but they smear together and have no discretely discernible attacks.

To get a single echo from the E1010, set the Feedback control at zero, and adjust the time delay as desired. For multiple echoes, turn up the Feedback control, If the input program has little or no sharp musical attacks (i.e., legato string lines or melodic vocal backups), then the multiple echoes with a moderate to long delay time can sound like a reverb.

Rhythmic Effects

Unusual rhythmic effects can be obtained by setting the E1010 for a particular delay, usually long, and playing against it. Thus, there is a direct interaction between the performer and the sound equipment. Whole musical pieces can be built around this interactive principle.

Vibrato

Vibrato, while it is a pitch change function, may be created with the E1010 analog delay line, and is based on the doppler shift phenomenon. It becomes a simple matter to add vibrato to voice and to instruments which cannot easily achieve the effect acoustically.

You can use the E1010 to vary the pitch manually by moving the Delay time control back and forth, or you can set the unit for automatic pitch variation by means of the Modulation Frequency and Depth controls.

Pitch Bend with Feedback

In contrast to vibrato, which is a gradual and linear variation in pitch, it is possible to obtain sequenced changes in pitch by using a long delay range, feedback, and gradually changing the delay time (manually or with the Modulation controls). A similar effect is obtained by moving the Delay control up or down and then holding the new setting; the pitch will increase or decrease to a new value and then hold at that value until the feedback dies out or a new sound is introduced.

Flanging

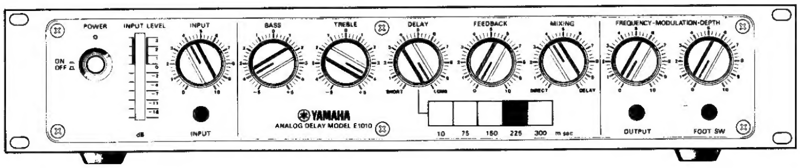
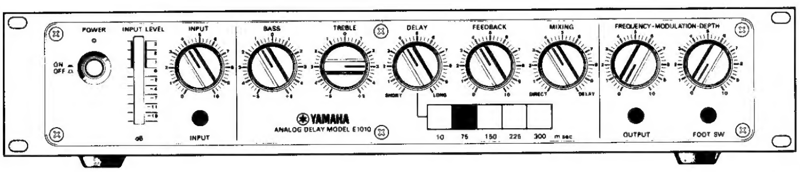
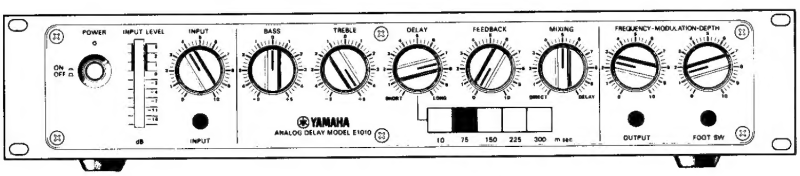
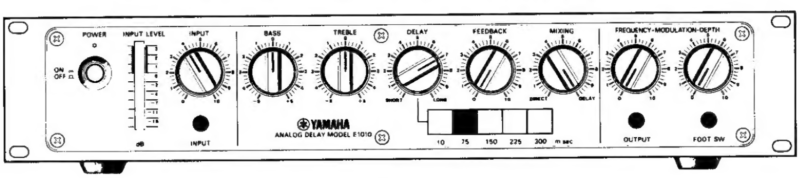
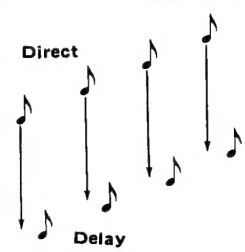
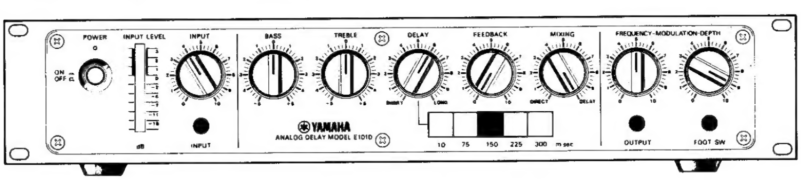
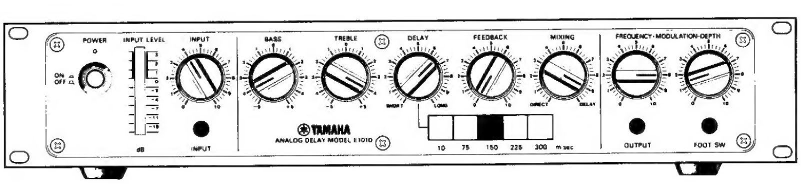
The term "flanging" comes from a practice known as "reel flanging." Two tape machines loaded with the exact same program are started in perfect sync; then alternately slowing down one tape machine and the other by pressing a hand against the flanges of the tape reels creates a series of harmonically related phase cancellations in the program. Unfortunately, true "reel flanging" is difficult to achieve – much less to repeat – as there is always the danger of completely losing tape machine synchronization. At best the technique is cumbersome because it requires the full attention of an operator other than the mixer or performer.

The E1010 creates flanging effects automatically, repeatably, and without the need for a single tape machine. To obtain the effect, set the unit for the shortest delay time, use *modulation* of the delay, and mix the direct and delayed sounds together; the varying delay signal beats against the direct signal to create phase cancellations at harmonics of the fundamental.

Hollow & Tunnel Effects

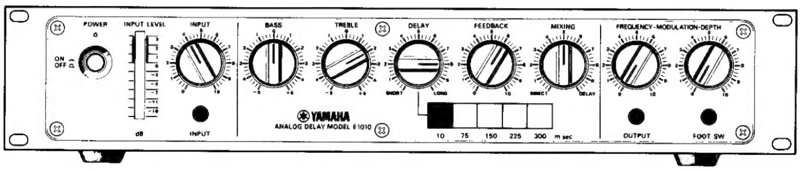
The E1010 can make normal voice or instrumental parts sound like they are coming from a hollow cavity or tunnel. This is done by setting the unit for a short delay, and by mixing the direct and delayed sound. This is similar to flanging, but there is *no modulation* so the result is a hollow-sounding comb filter. If the delay time is increased slightly and feedback is added, then more of a tunnel-like sound (flutter-echo) is obtained.

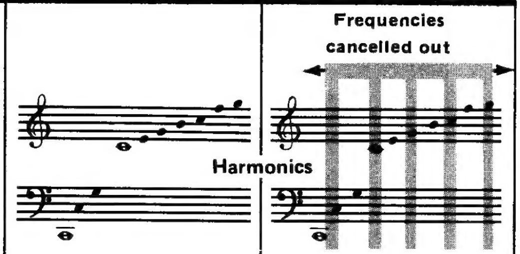
Fig. 7 — E-1010 Front Panel Settings & Musical Representations of Various Effects.
NOTE: Those controls illustrated without specific settings may be adjusted as necessary without changing the basic effect.

	INPUT	OUTPUT
ECHO *For multiple echo, set "Feedback" at 6.* 		
REVERB 		
DOUBLE TRACKING 		
MONO-TO-STEREO *Must use rear panel outputs, each feeding different channel.* 		
AUTOMATIC PITCH BEND *For manual pitch bend, set "Modulation" controls at 0 and move "Delay" while playing.* 		
VIBRATO 		

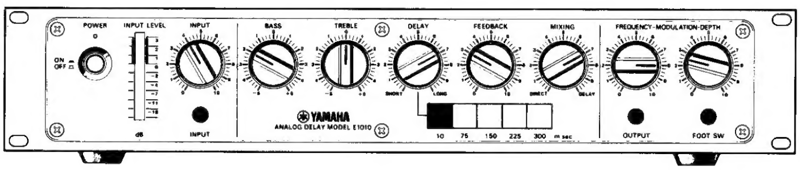
INPUT	OUTPUT

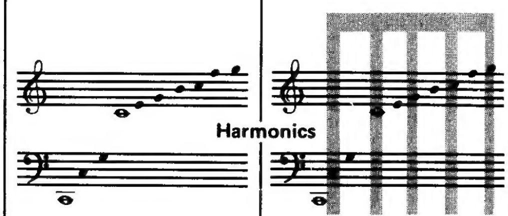
FLANGING



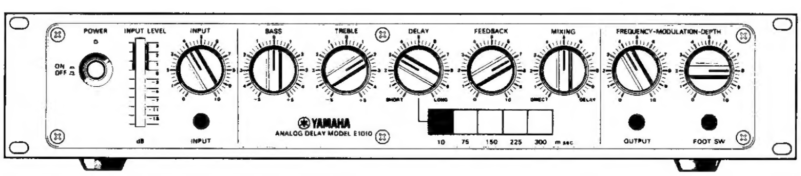


TUNNEL





PHASE SHIFTING



DISTRIBUTED SPEAKER SYSTEMS & THE HASS EFFECT

While the E1010 is designed primarily for creative signal processing, it can be used in conventional delay line applications. For example, when an extension speaker is placed remotely from a main stage speaker, the sound may be perceived as coming from the remote speaker and not the stage, depending on the relative distances between the listener and the speakers. With the E1010 one can ensure that sound is perceived as coming from the main speakers.

If the remote audience loudspeaker is 25 or more feet from the main stage speaker, a listener near the remote speaker will either not hear the main speaker or will hear it as the source of an echo. However, if the E1010 is used to delay the signal to the *remote* speaker* so that its sound actually arrives at the audience 5 to 20ms later than the sound from the *main* speaker, then the perceived localization of the sound will be from the stage; the remote speaker still reinforces the sound from the stage, but is not heard as a distinct sound source. Our localization of a sound "image" based on the sound source which arrives first at our ears is known as the Hass effect. (Refer to Figure 8 for a graph of delay times, speaker-listener distances, etc.).

In using the Hass effect and delaying the sound to the remote speaker(s) with the E1010, the delay line can actually eliminate echoes rather than create them.

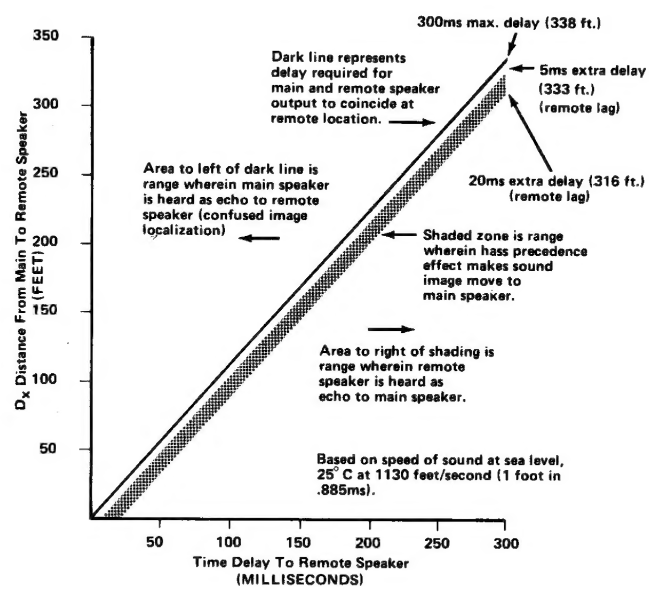


Fig. 8 — Delay Times required for hass effect correction of sound images at various main speaker/remote speaker/listener distances.

**The E1010 must operate on line-level signals fed to the remote power amplifier, not on the amplifier output.*

A SIMPLIFIED FUNCTIONAL DESCRIPTION

You don't have to know how the E1010 works in order to use it. However, a careful reading of this section will enable you to get the most out of the unit. Instead of guessing, you can truly understand how the E1010 processes the sound, predict the sound obtained at various control settings *before* you even turn the unit ON, and duplicate effects without having to document the settings. The following paragraphs are numbered to correspond with the blocks in Figure 9, which depicts basic signal flow through the unit.

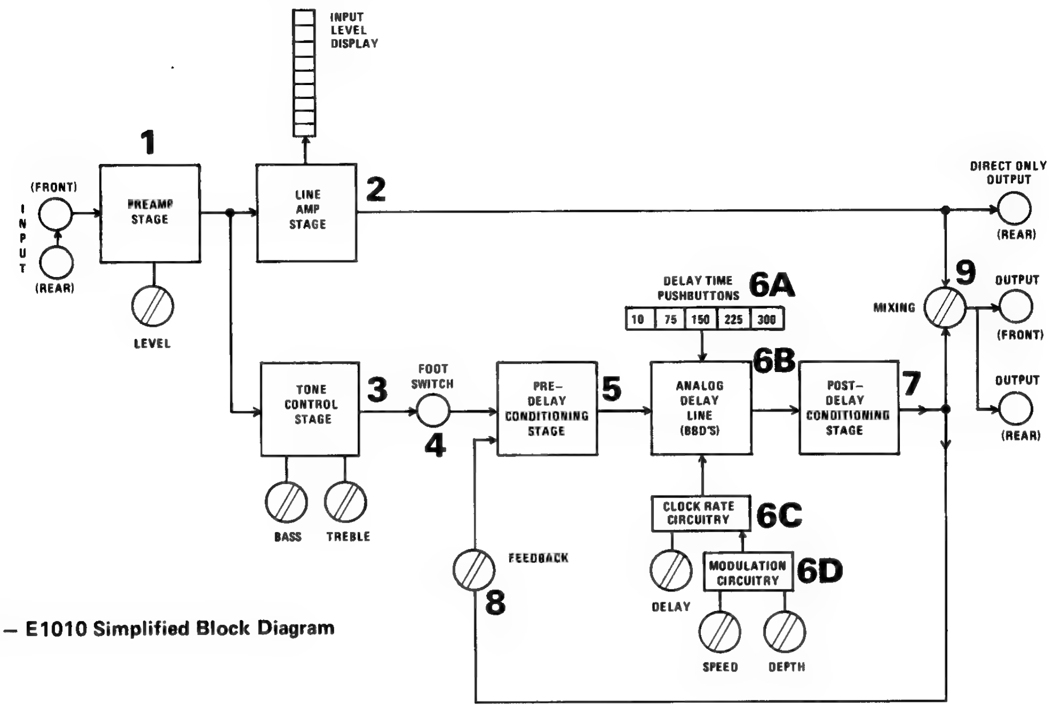


Fig. 9 — E1010 Simplified Block Diagram

Preamplifier Stage (1)

The input signal from either the front or rear panel INPUT jack goes through two preamplifiers, one on either side of the INPUT volume control. Because the INPUT control is buffered (comes after a stage of amplification), volume adjustments do not change the input impedance; hence the tone from a high impedance guitar pickup would not shift with volume adjustments. After the preamp stages, the signal is split, part of it going to the DIRECT OUTPUT, and part going to the E1010's tone control and delay circuitry.

Direct Output Path (2)

The signal goes through a Line Amplifier before it reaches the DIRECT OUTPUT. The INPUT LEVEL display circuits sense the voltage at this point and cause more LED's to turn ON as the level increases. Thus, only the VOLUME control affects the INPUT LEVEL display.

Tone Control Stage (3)

The Tone Control Stage is an equalizer which has BASS and TREBLE controls. Any tone adjustments only affect the delayed portion of the signal.

Foot Switch Jack (4)

The FOOT SWITCH JACK comes between the Tone Control Stage and the Delay circuits. It permits the input to the delay circuits to be shorted (turned off) if an external foot switch is plugged in. It is thus possible to turn the delay effect ON and OFF without affecting the E1010's direct output signal. When no switch is used, the signal goes straight through the jack so the delay is ON.

Pre-Delay Conditioning (5)

This circuitry prepares the audio for delay by cutting out very high frequencies (via a low pass filter), boosting mid to high frequencies (pre-emphasis), and preventing the signal from getting too loud or soft (compression). The pre-delay processing helps to avoid noise, distortion, or overload of the actual Delay circuitry (6).

NOTE: Up to this point we have described circuitry that, while necessary, is not directly involved with delaying the audio. The actual "delay line" itself (6A, 6B, 6C and 6D) comprises only a portion of the E1010's overall circuitry.

Analog Delay Circuitry (6)

The Delay Circuitry consists of several integrated circuits (IC "chips") of the variety known as BBD's (Bucket Brigade Devices). Each BBD provides a variable amount of delay, up to a certain maximum; more BBD's can be connected together to increase the maximum available delay.

A. The Delay Time pushbuttons determine how many of the BBD's are connected in the signal path; a 300 millisecond delay utilizes all the BBD's.

B. The BBD's slow down the audio signal by dividing it into small segments, and then passing those segments through a very long series of storage registers. Each signal segment stays in a storage register for a brief period before moving on and making room for the next signal segment to take its place. The length of time a signal resides in a storage register is determined by the Delay Time Adjustment circuitry, which is why engaging any

given Delay Time pushbutton (6A) can actually produce a wide range of delay times.

C. The Clock Rate Circuitry sets the speed at which the audio signal segments move from one storage register to the next. The clock is simply a variable oscillator whose frequency may be changed by adjusting a control voltage (i.e., a VCO — voltage controlled oscillator). Higher settings of the DELAY control slow down the clock which in turn slows down the movement of audio through the BBD's, and hence increases the delay time. However, the BBD's can only be slowed down by a finite amount before they cease to function properly; longer delays then require the use of additional BBD's, which is why the Delay Time pushbuttons are necessary.

NOTE: The process is analogous to moving a large water tank (the musical program) from one point (the input) to another (the output) by pouring it along a series of buckets (the BBD's)—hence, the term "delay line." The time it takes to move the water from one tank to the other depends on how many buckets are used (the Delay Time) and how fast the water is transferred from one bucket to the next (the Clock Rate circuitry).

D. The Delay Time MODULATION circuitry really does the same thing as the DELAY control; it changes the Clock Rate. The only difference is that the Delay Time Modulation circuitry automatically varies the clock rate. The DEPTH control sets the maximum change in clock rate (and hence the maximum change in delay time). The FREQUENCY control sets the speed at which the clock rate is varied (and hence the rapidity of changes in delay time).

Post-Delay Conditioning (7)

This circuitry restores the original balance and dynamics by doing the reverse of the Pre-Delay Conditioning (5). The previously boosted mid to high frequencies are cut, thereby duplicating the original frequency response while simultaneously reducing any noise in this range which might have been introduced by the Delay circuitry (de-emphasis). The loud passages are made louder and the soft passages made softer, hence restoring the original dynamic range and simultaneously forcing low-level noise to even lower levels (expansion). Very high frequencies are cut out, thereby avoiding any clock noise which might have been introduced in the Delay circuitry (low pass filtering). At the output of this stage, the audio sounds like it did before being delayed, only it is offset in time.

The Feedback Path (8)

The FEEDBACK control permits a portion of the delayed signal to be re-applied to the delay circuitry. This creates a loop which is useful for special effects. At longer delay times, the feedback is heard as repetitive echoes. At shorter delay times, the feedback creates a series of signal cancellations which are head as "flanging," "comb filtering," "tunneling," etc. At very high FEEDBACK settings, the entire delay line can go into self-oscillation, even without an input signal, and create unwanted howling.

Direct/Delay Signal Mixing (9)

The MIXING control affects the front- and rear-panels OUTPUT. It determines the proportion of direct and delayed sound fed to the output. Due to phase cancellations, it is possible to achieve certain effects, effects which could not be achieved by feeding direct and delayed sounds to separate amplifier/speaker systems, by electrically mixing direct and delayed sound together in that one output.

A CLOSER EXAMINATION OF THE DELAY CIRCUITRY

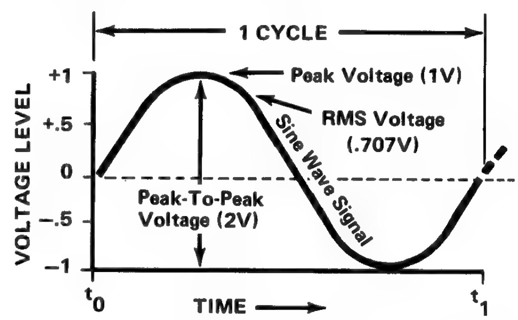
General Discussion

The bucket brigade devices (BBD's) and the clock (oscillator) constitute the heart of the analog delay line. All the other circuits (tone control, pre- and post-delay conditioning, feedback and mixing) are of secondary importance; in fact, similar circuits to these are also used in digital delay lines. The following paragraphs explain how the signal is delayed inside the E1010, why higher frequencies are lost at longer delay times, and how the analog delay technique differs from digital delay techniques.

For the purpose of this discussion, assume the input is one cycle of a simple sine — a brief burst of pure tone. Any signal could be used, but this one is easy to visualize. The original audio program (an analog signal) is nothing more or less than a voltage level which varies up and down. A graph of instantaneous voltage level (vertical axis) versus time (horizontal axis) gives the familiar representation of audio waveform shown in Figure 10.

Sample and Hold: The Key to Analog Storage

As the signal is applied to the first BBD, it is divided into small segments of equal length, creating a continuous stream of program samples. The exact length of each sample is defined by the E1010's clock pulses, and each sample has an average voltage value that is taken directly from the signal present during the corresponding clock pulse. (Refer to Figure 11) These samples are stored (held) to create the time delay.



If $t_1 - t_0$ = 0.5 milliseconds, frequency = 2,000Hz (2kHz),
If $t_1 - t_0$ = 1.0 milliseconds, frequency = 1,000Hz (1kHz),
If $t_1 - t_0$ = 2.0 milliseconds, frequency = 500Hz (.5kHz),
If $t_1 - t_0$ = 1 second, frequency = 1Hz, etc.

Fig. 10 — A Simple Sine Wave Audio Signal

Reconstructing the Original Waveform

After a sufficient number of clock pulses, the samples begin to exit from the last storage register of the BBD. At this point, a waveform is reconstructed. It is very similar to the original input signal, but rather than being a smooth and continuous wave, it moves up and down in steps, corresponding to the average voltages of the samples. The steps can be considered to be a high frequency noise (clock noise) superimposed on the original waveform. Thus, by routing the signal through a low pass filter, the steps are eliminated and the waveform is smoothed out to be very nearly identical to the original input signal, except that it is present at some later time (i.e., it is delayed). (Refer to Figure 11.)

Fig. 11 — The Analog Waveform is divided into a series of samples whose corresponding average voltages move through storage registers in a bucket brigade device (4096 Storage Registers in the BBD shown).

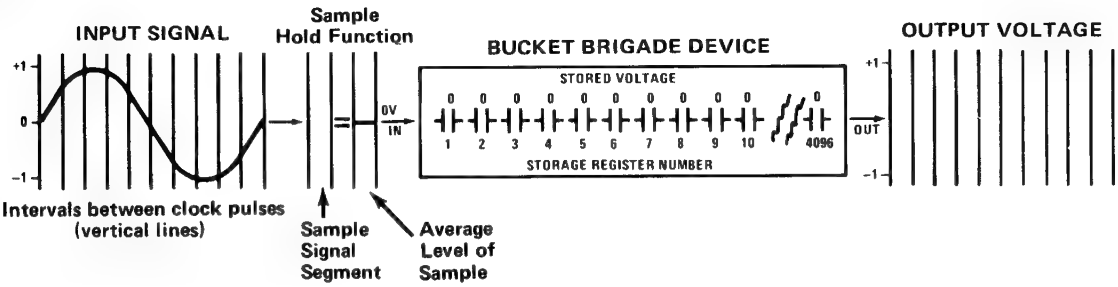
WHAT IS A STORAGE REGISTER?

It is a cell similar to a capacitor, a device capable of holding a voltage for an indefinite period of time (after a long time, the charge will leak away unless it is transferred to the next storage register). The various signal samples (voltages) thus move through the in storage registers without changing their values.

T = 0

Prior to the first clock pulse, it is assumed there is no signal in storage, so all registers have 0 volts present.

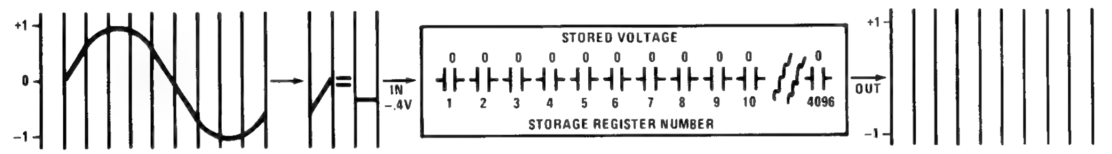
T = 0



T = 1

This first clock pulse marks the beginning of the # 1 signal sample.

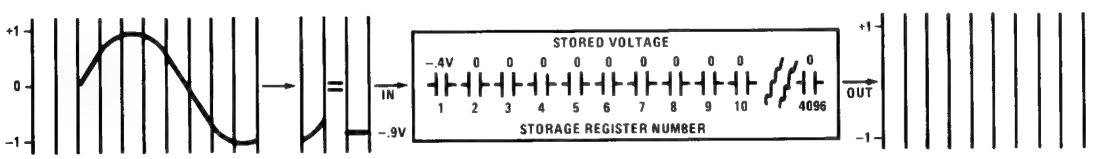
T = 1



T = 2

The next clock pulse marks the end of the #1 sample and the beginning of the #2 sample. At this instant, the #1 sample enters the first of many storage registers.

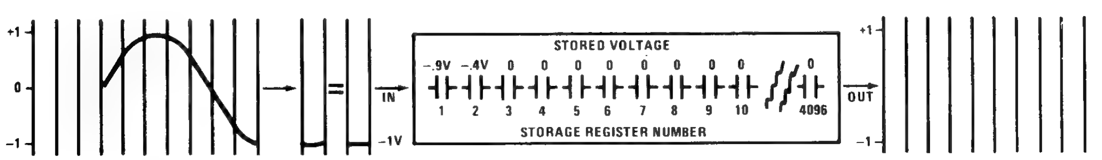
T = 2



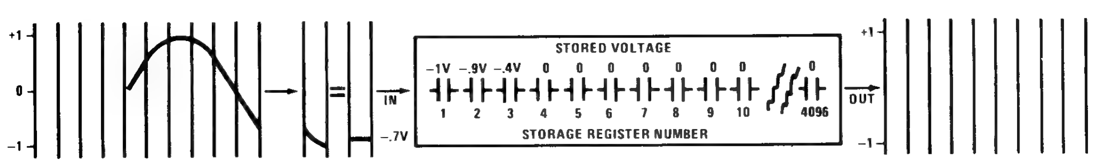
T = 3 to T = 11

At the next clock pulse, the #1 sample moves out of the first storage register and into the next one, making room for the #2 pulse to enter the first register. Simultaneously, the #3 sample is being taken. This process continues with each clock pulse defining a new signal sample and pushing previously defined samples into subsequent storage registers until the entire signal is stored in the BBD.

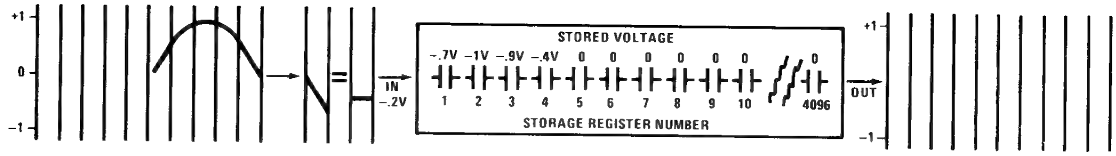
T = 3



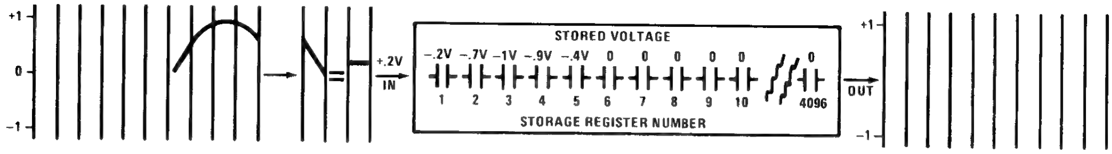
T = 4



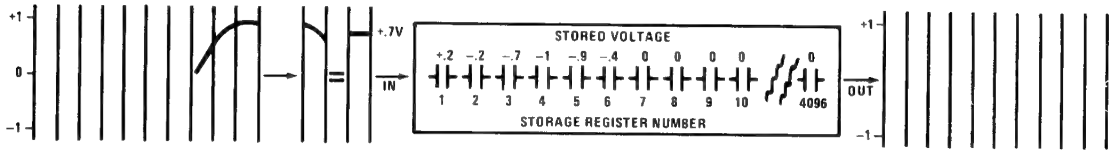
T = 5



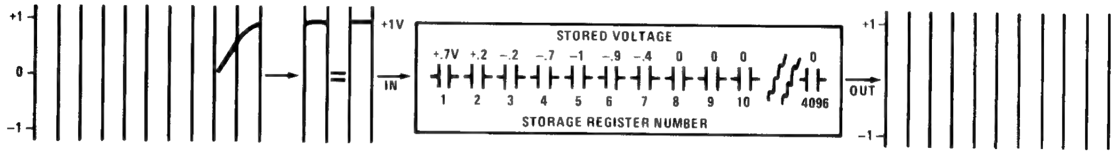
T = 6



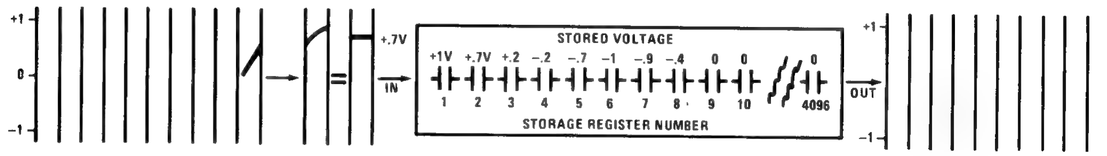
T = 7



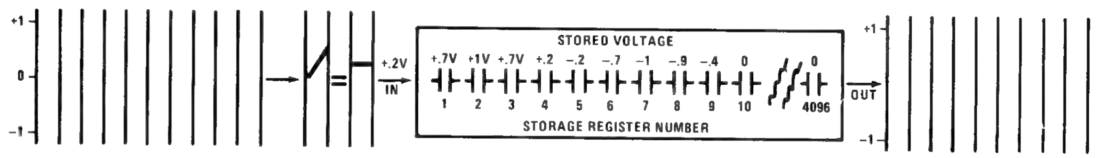
T = 8



T = 9



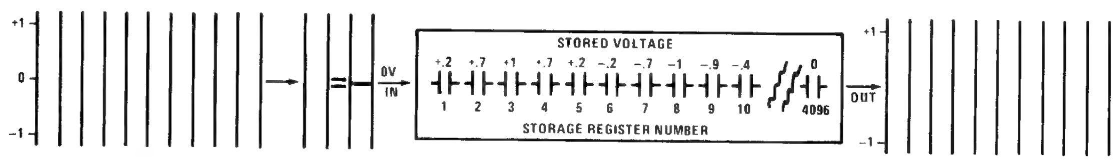
T = 10



T = 11 to T = 4095

The samples move serially through the BBD storage registers and begin to appear at the BBD's last storage register, ready for output as voltages.

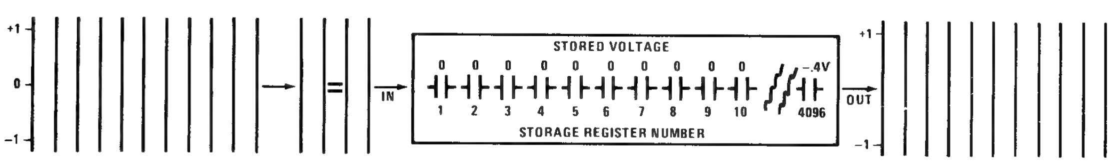
T = 11

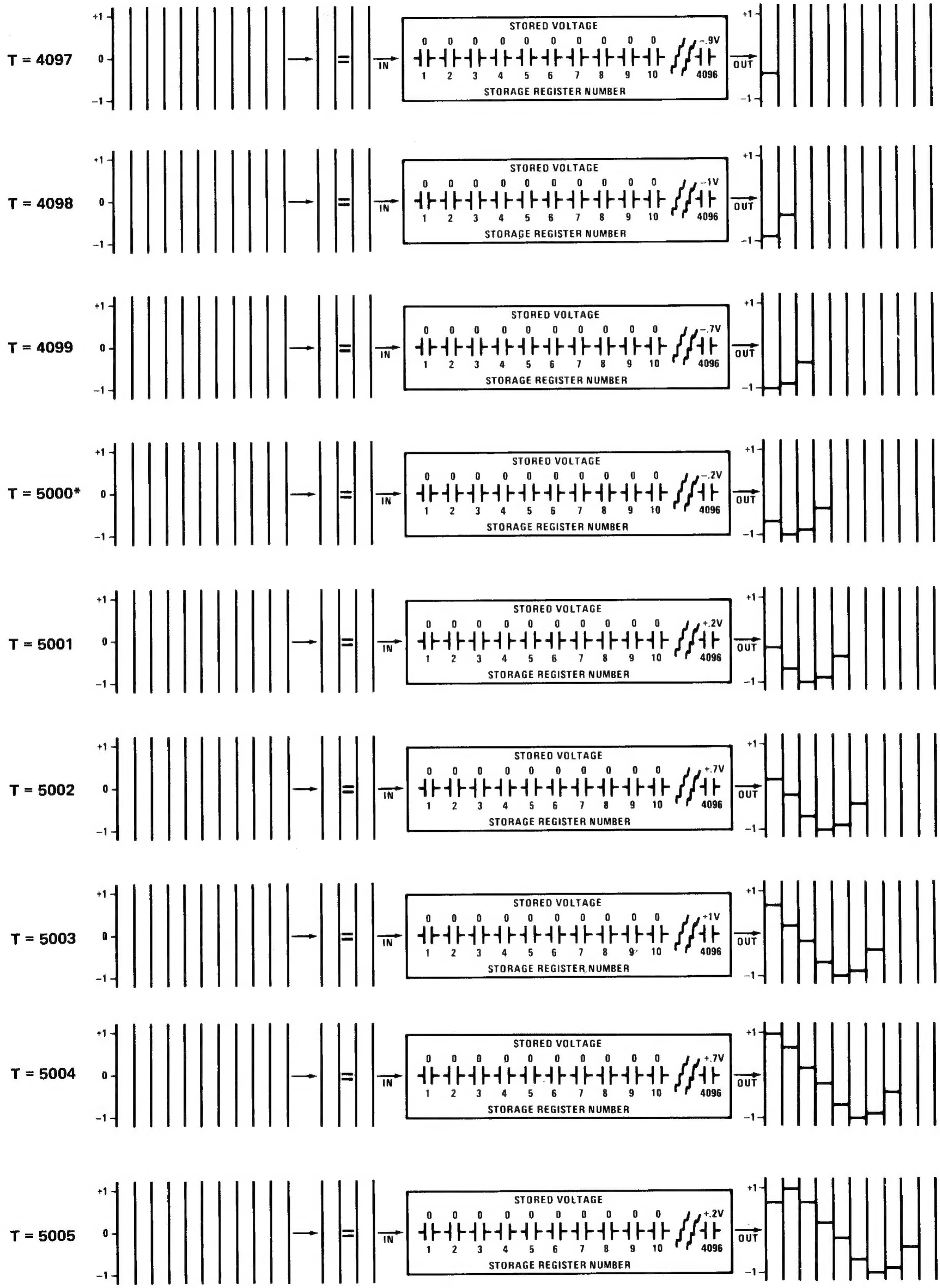


T = 4096 to T = 5006

The samples continue moving out of the BBD until the entire signal has been output as a rough waveform, a string of discrete voltage steps.

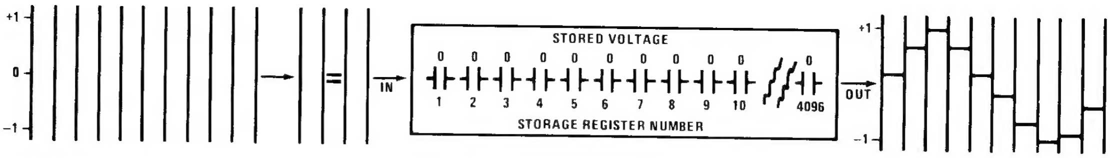
T = 4096





**NOTE: At a clock rate of 50kHz, T5000 equals 1/10 second of delay (100 milliseconds).*

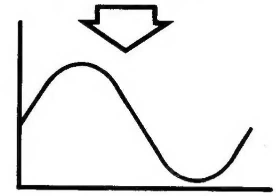
T = 5006



As the samples exit the BBD, they go through a Low Pass Filter which smooths them and restores the original signal. In this illustration, the output appears approximately 100mS after it was input.

NOTE: These samples are very coarse for convenience of illustration; there would normally be many more samples per cycle of audio.

LOW PASS FILTER



Output Signal

A Comparison of Analog and Digital Delay Lines

A digital delay line (DDL) generally incorporates signal conditioning circuitry which is similar to an analog delay line (ADL), but has another circuit element, an analog-to-digital (A-D) converter. The A-D converter takes each sample of the waveform and gives it a numerical value to represent its voltage level and polarity (+ or –). These numbers are stored in random access memory (RAM) or in digital shift registers instead of bucket brigade devices. After the desired delay, the numbers are retrieved from memory, go through D-A converters and again become voltage levels. These voltages are rough waveforms and, like the analog delay's BBD output, they must be low pass filtered.

The A-D and D-A converters add to the cost of the digital delay line and provide an additional source of noise known as "quantizing noise." The ability of any DDL to handle wide dynamic range and frequency response depends largely on how many digits are used for each number —each stored voltage value. 12-bit (12 digit) DDL's are cheaper, but less desirable than 14-bit, 16-bit or higher resolution DDL's. 12 or more bits may seem like a lot, but remember this is a binary number (base 2), and the polarity, direction of voltage swing, and actual value must all be documented. Similar effects and delay times can be achieved in DDL's and ADL's. The major differences are in cost and performance. The most sophisticated DDL's can yield better audio performance than typical analog delays, but the cost of a DDL is considerably higher than an ADL of comparable audio quality and maximum delay time.

Frequency Response versus Delay Time

If a longer delay is desired, the clock rate may be slowed down (the E1010 DELAY control is turned up). However, as the clock is slowed down, the duration of each sample time increases, so a larger proportion of

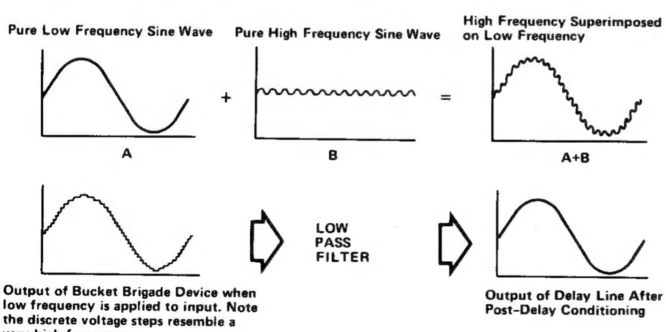


Fig. 12 — High frequencies "Ride" on lower ones and may be filtered out to "Smooth" waveform.

the input signal falls into each sample. From Figure 12, it can be seen that high frequencies actually "ride" on low frequencies; they are the smaller "squiggles" superimposed on the dominant waveform. When a larger (coarser) sample is averaged, any small variations within the sample are lost, hence the high frequencies are lost. Thus, slowing the clock rate, while it does lengthen the delay time, also cuts the frequency bandwidth.

Another way to increase the delay time maintains the same clock rate but utilizes more storage registers. This is exactly what occurs when the E1010 Delay pushbuttons are switched to a longer delay range. At first it might seem that this technique would provide longer delays with no loss of high frequencies. Due to the way the BBD's function, it is necessary to insert an additional low pass filter with each additional BBD, so there is still some high frequency loss as the delay time increases. However, the signal conditioning (pre-emphasis and de-emphasis) helps to minimize high frequency roll-off.

HOW FAST DOES THE CLOCK RUN?

The number of clock pulses per second constitute its rate (frequency). The clock, an oscillator, must be at least twice as fast as the highest audio frequencies one wishes to delay. In real numbers, a 50kHz clock rate would be desirable for a 20kHz upper audio response limit, 30kHz for a 13kHz upper limit, etc. This is why the high frequency response falls off as the clock is slowed down to achieve longer delays. Extremely high clock rates are generally avoided, even though they could provide better high frequency response, because (a) more BBD's are required to obtain a given delay, and (b) the related high-speed circuitry is more complex and costly.

How Delay Time Changes Can Produce Pitch Changes

Pitch (frequency) is determined by how many waves occur in a given unit of time, i.e. cycles per second. For instance, a 1kHz signal (1,000 cycles/second) is the same as 1 cycle per millisecond. Normally, any signal applied to the E1010 input is exactly duplicated at its output, but is merely offset in time by a given delay. Thus, one cycle of a 1kHz sine wave would take 1 millisecond to emerge from the output jack of the E1010, but it might come out as much as 300 milliseconds (300mS) after it entered the unit's input.

If the delay time is decreased while a signal is coming out of the E1010, then the pitch will be decreased— and vice-versa. For example, if the E1010 delay time is turned down from 300mS to 150mS just as the aforementioned 1kHz wave is emerging, then the wave will

take 2mS (not 1mS) to come out of the unit. Hence, its pitch is cut in half to 500Hz. At this point, if another 1kHz wave is applied to the E1010 and no further delay time adjustments are made, it will again take 1mS for the wave to come out of the unit — still a 1kHz wave, but emerging 150mS after it entered. If the delay time is then increased from 150mS back to 300mS just as another 1kHz wave is emerging from the E1010, that wave will take only 1/2mS to come out of the unit. Hence, its pitch is doubled to 2kHz. (Refer to Fig. 13.)

How Vibrato is Obtained

In the preceeding paragraphs we explained how pitch is changed by changing the delay time. If the DELAY control were turned up and down in a regular manner, a vibrato (rhythmic pitch change) would be produced. Try it. The E1010's MODULATION circuitry provides for automatic changes in delay time; DEPTH sets the amount of change and FREQUENCY sets the speed of the change.

How Lower and Higher Pitches Can Be Sustained Together

Consider how and when the E1010 changes the pitch of a sound. The pitch of a sound changes only when it is emerging from the E1010 output as the delay time is being changed. As soon as a new delay setting is fixed, the pitch of new sounds will not be changed. However, there is a way to preserve the changed pitch which occurred as the delay was changing — use FEEDBACK.

Suppose the FEEDBACK control is turned up and a sound is applied to the E1010 input. Simultaneously, the DELAY control is moved up or down to change the pitch of a sound, and the control is quickly released. What happens? The pitch-changed output re-circulates through the delay unit, sustaining whatever pitch was reached when the delay time stopped changing. Now what happens if a new "normal pitch" input is applied? It goes through the delay and maintains its normal pitch, mixes with the sustained pitch-changed sound, and the two sounds sustain together until the feedback dies away. With a little practice, it is possible to play or sing the same note three times in quick succession, move the DELAY control during the first two notes, and actually create a triad.

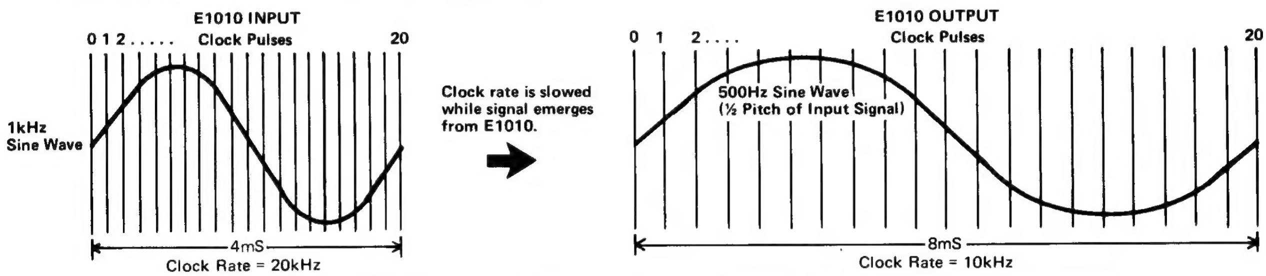
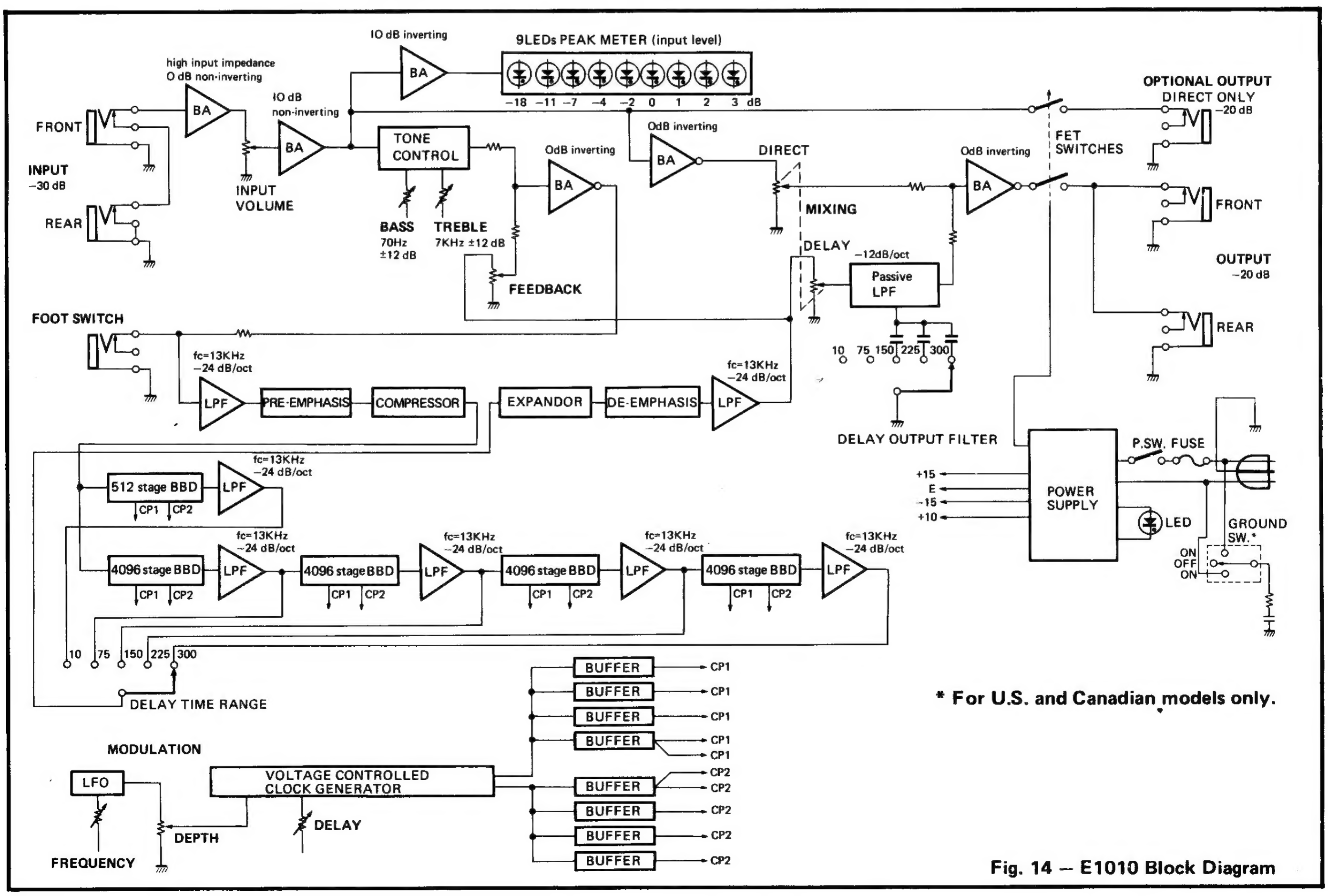
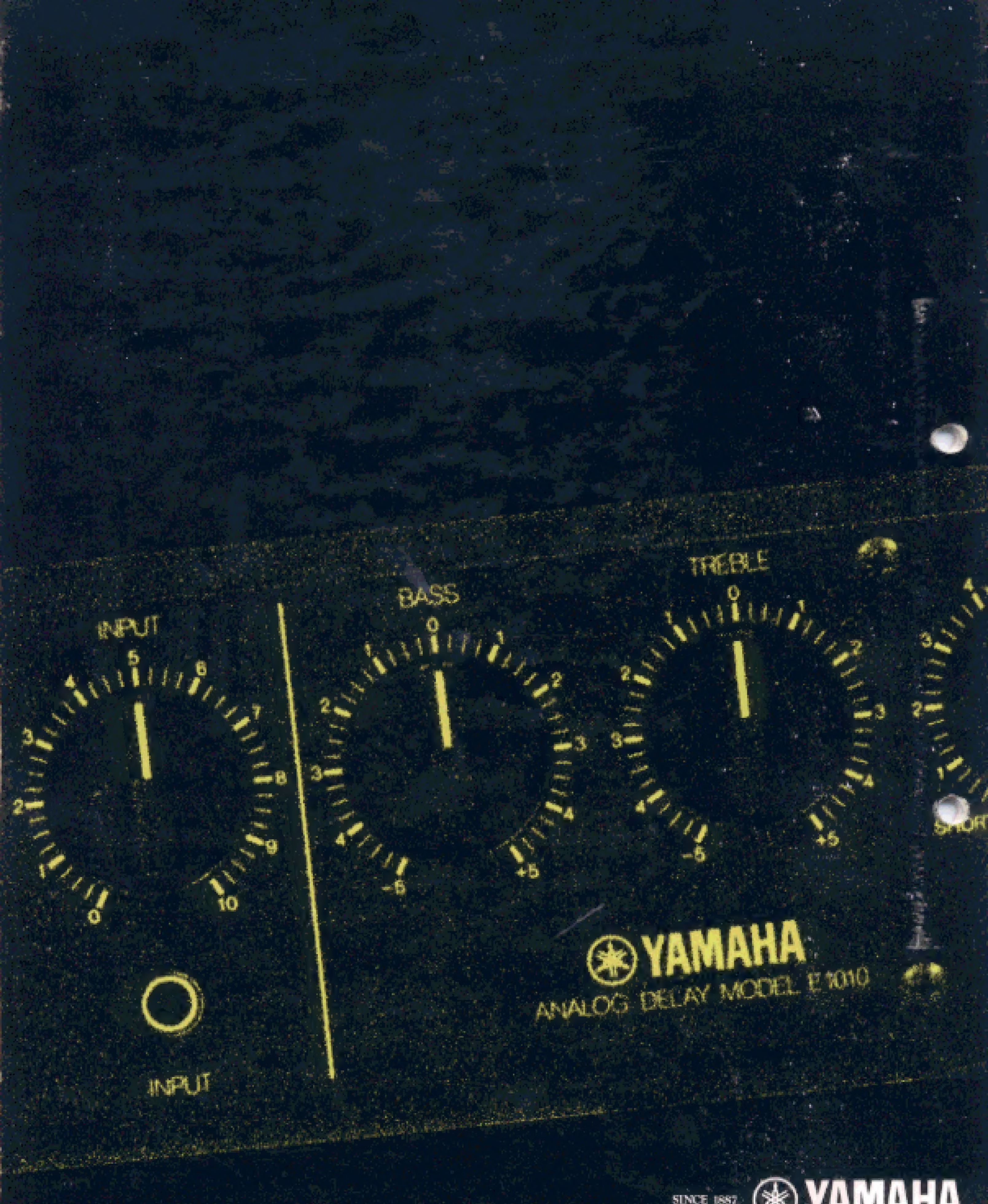


Fig. 13 — How changing the clock rate as a signal emerges from the E1010 changes the pitch.



* For U.S. and Canadian models only.

Fig. 14 — E1010 Block Diagram



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